COMMUNICATION NETWORK AND COMMUNICATION SYSTEM FOR

IMPLEMENTING THE METHOD--;

line 7, insert --TECHNICAL FIELD OF THE INVENTION--.

line 12, delete "especially" and replace with --and in particular to--.

between lines 16 and 17, insert --BACKGROUND OF THE INVENTION--.

line 19, insert --,-- after "network".

line 19, delete ", for example,".

line 20, delete "so called".

lines 20 and 21, delete "the current aim is" and replace with --it has become increasingly popular--.

line 21, delete "increasingly".

line 24, delete "network" and replace with --networks--.

Page 2,   between lines 2 and 3, insert --SUMMARY OF THE INVENTION--.

Page 2,

directly after "SUMMARY OF THE INVENTION", please insert the following:

--   In one embodiment of the invention, there is a method for improving the quality of an audio transmission in which audio data including samples of an audio signal are asynchronously transmitted in data packets from a transmitting communication system via a packet-oriented communication network to a receiving communication system. An information item relating to the transmission of data packets is detected, wherein the audio data are converted such that their sampling rate is altered by means of digital filtering, the sampling rate being altered based on the detected information item, in such a manner that due to the altered sampling rate, a quality of service of the audio transmission is optimized with regard to a current transmission situation indicated by the detected information item.

In another embodiment of the invention, there is a method for improving the quality of an audio transmission in which audio data including samples of an audio signal are asynchronously transmitted in data packets from a transmitting communication system via a packet-oriented communication network to a receiving communication system. An information item relating to the transmission of data packets is detected, wherein the audio data are digitally converted such that the duration of an audio signal represented by the audio data is modified while retaining a pitch of the audio signal, the duration being modified based on the detected information item, in such a manner that due to the modified duration, a quality of service of the audio transmission is optimized with regard to a current transmission situation indicated by the detected information item.

In one aspect of the invention, the audio data to be transmitted are converted by the transmitting communication system and a conversion message about the conversion is transmitted from the transmitting communication system to the receiving communication system.

In another aspect of the invention, the transmitted audio data are reconverted by the receiving communication system, the change in the audio data taking place in the reconversion being determined by means of the conversion message transmitted.

In still another aspect of the invention, the transmission of the data packets is monitored by the receiving communication system and an information item relating to this transmission is transmitted to the transmitting communication system and the audio data are converted by the transmitting communication system based on the information item transmitted.

In yet another aspect of the invention, the information item transmitted specifies a data packet loss rate and, if the data packet loss rate rises, the audio data are converted by the transmitting communication system in such a manner that the audio data rate is reduced.

In still another aspect of the invention, a detected incorrect adaptation of the data rate of the received audio data is at least partially compensated by the receiving communication system by means of a conversion of the received audio data.

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In yet another aspect of the invention, the received audio data are converted after having been read out of an input buffer provided for compensating data packet delay variations, in which the read-out speed of the input buffer is controlled by a change in the audio data rate due to the conversion.

In another aspect of the invention, the audio data included in the data packet preceding and/or following the lost data packet are converted by the receiving communication system such that the duration of an audio signal represented by the audio data is extended, in such a manner that the extension of the duration at least shortens a gap in the audio signal due to the lost data packet.

In still another embodiment of the invention, there is a communication system for transmitting and/or receiving audio data including samples of an audio signal via a packet-oriented communication network. The system includes, for example, a monitoring unit for detecting an information item relating to the transmission of data packets including audio data, a digital sampling rate conversion device for converting the audio data by altering their sampling rate and a control unit for controlling the sampling rate alteration based on the information item detected.

In yet another embodiment of the invention, there is a communication system for transmitting and/or receiving audio data including samples of an audio signal via a packet-oriented communication network. The system includes, for example, a monitoring unit for detecting an information item relating to the transmission of data packets containing audio data, a digital timescale conversion device for converting the audio data by changing the duration of an audio signal represented by the audio data while retaining a pitch of the audio signal, and a control unit for controlling the change in duration based on the information item detected.

In one aspect of the invention, the digital sampling rate conversion device exhibits a digital filter chip for converting the audio data.

Q2   In another aspect of the invention, the digital timescale conversion device exhibits a digital signal processor for converting the audio data.--

Page 2,   line 3, delete, "It is the object" and insert --In one embodiment--.

lines 3 and 4, delete "present invention to specify" and replace with --invention, there is--.

line 6, delete "It is" and replace with --There is also--.

line 7, delete "further object to specify".

Please delete page 2, lines 9-38.

Please delete page 3, lines 1-38.

Please delete page 4, lines 1-38.

Please delete page 5, lines 1-3.

Page 5   between lines 3 and 4 insert --BRIEF DESCRIPTION OF THE DRAWINGS--.

Page 5   between lines 15 and 16, insert --DETAILED DESCRIPTION OF THE INVENTION--.

Page 5   directly after "DETAILED DESCRIPTION OF THE INVENTION", insert the following:

-- The quality of audio transmissions via any packet-oriented communication networks such as, for example, local area networks (LANs) or wide area networks (WANs) can be improved in a simple manner by means of the invention. The invention can be used advantageously, in particular, in packet-oriented communication networks which do not provide a guaranteed quality of service (QoS). Since it is not necessary to intervene in an existing communication network to be used for transporting the audio data, most of the existing packet-oriented communication networks can be used with the invention.

According to the invention, the quality of an audio transmission is improved by regulating the audio data rate in dependence on the respective transmission situation. The audio data rate is changed by a digital conversion of the audio data. In this arrangement, the audio data is converted in the sense of an alteration of their sampling rate, i.e. the samples of the audio signal produced per unit time, and/or in the sense of a modification of the duration of an audio signal represented by the audio data while largely maintaining its pitch. The first type of conversion mentioned is called "sample rate alteration" (SRA) and can be performed in a simple manner, for example with digital filter chips. The second type of conversion mentioned is called "time scale modification". Various algorithms for performing this conversion are described, for example, in "Time-Scale Modification of Speech Based on Short-Time Fourier Analysis" by M.R. Portnoff, IEEE Transactions on ASSP, July 1981, pages 374 to 390, in "Shape Invariant Time-Scale and Pitch Modification of Speech" by T.F. Quatieri and R.J. McAulay, IEEE Transactions on Signal Processing, March 1992, pages 497 to 510, and in MPEG-4 Audio, ISO/IEC FCD 14496-3 subpart 1, section 4.1, dated 5.15.98.

The audio data rate can be altered within wide limits and regulated more precisely by the two aforementioned conversion methods than with previous data compression methods normally used in conjunction with packet-oriented audio transmissions. Both conversion methods allow continuous audio data streams to be converted and only delay these to a minimum extent, which results in very good real-time characteristics.

The conversion methods can be implemented both individually or in combination with each other in a given communication system transmitting the audio data and/or in a communication system receiving the audio data. Exemplary communication systems include, for example, audio terminals, audio switching systems such as, for example, so-called private branch exchanges (PBXs) and, in particular, gateways and clients according to ITU-T Recommendation H.323 of the International Telecommunication Union.

According to an advantageous embodiment of the invention, the audio data to be transmitted can be converted by the transmitting communication system and a conversion message relating to the conversion can be transmitted to the receiving communication system. The transmitted conversion message can then be used by the receiving communication system for controlling a reconversion of the audio data. The conversion of the audio data performed at the transmitter end can be largely cancelled again, for example, by a reconversion at the receiver end so that the audio signal represented by the audio data is equalized again.

Transmission of the audio data can also be monitored by the receiving communication system and an information item relating to the transmission can be transmitted to the transmitting communication system. This can then convert the audio data in dependence on the transmitted information item. Thus, for example, the audio data rate, and the data packet rate, can be reduced by a conversion at the transmitter end if the receiving communication system reports an increasing data packet loss rate.

The invention can also be advantageously used for synchronizing communication systems. This can be achieved by converting the received audio data after reading them out of an input buffer provided for equalizing data packet delay variations. In this case, the read-out speed of the input buffer can be controlled by controlling the conversion ratio of the audio data rate as a result of the incorrect synchronization for example, a so-called "delay jitter" can be compensated.

According to an advantageous further development of the invention, packet losses can be compensated by the receiving communication system by extending in time a data packet preceding and/or following a lost data packet by a conversion according to the invention in such a manner that a gap in the audio signal due to the lost data packet is closed or shortened.

Furthermore, the data rate of audio data to be transmitted can be lowered by a conversion in favor of a transmission of additional redundancy information such as, for example, error correction bits and/or CRC check information.

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